For Quick Compression - Sound attached to it called pumping sound.
(This doesn't happen in adaptive compression)

Directional Mic. | Omni (all directions) |
Single direction

DIR microphone
One port through which the sound comes in → Diaphragm (vibrates) → Electrical energy.
The mic will create an ~~analo~~ electrical signal analogous (similar face, amplitude, time) to acoustic signal.

DIR Mic.
HAS effectively reduce noise and ↑ speach because of DIR mic. (thus improve SNR)
Has 2 inlets.
Sound reaches port ① & then port ②
Therefore, time lag in port ②. This delay is called External delay (Happens outside the Mic)
The mechanical screen will artificially delay the signal through port ① so that both signals come at the same time.

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Both sounds come in opp. direction. So they cancel each other & diaphragm won't vibrate. So all rear sounds will not be heard, only front sounds. All sounds from rearside/ backside is cancelled out.

Sound has to travel from Rearside to Frontside, so there is an external delay

(1 microphone with 2 ports)

Current day.

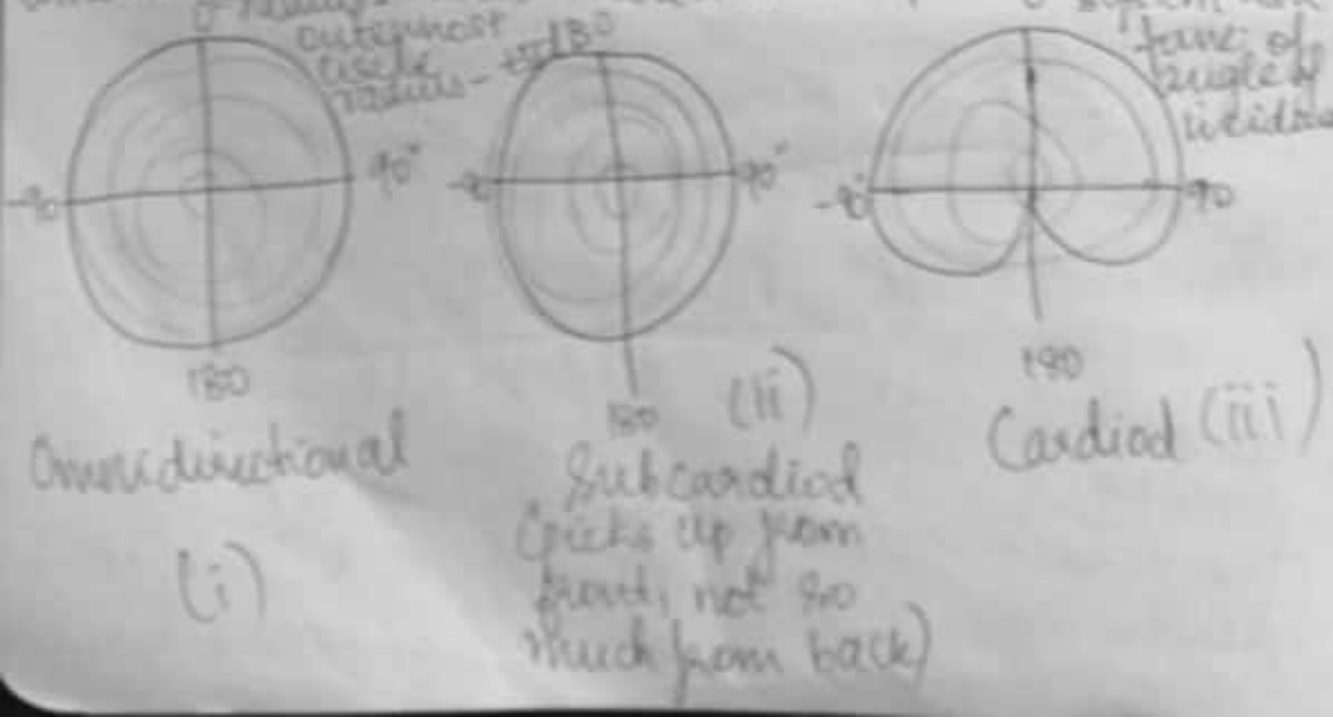
Twin microphone (2 microphones)

How does it achieve directionality?

Draw it.

To visually demonstrate the directivity pattern.

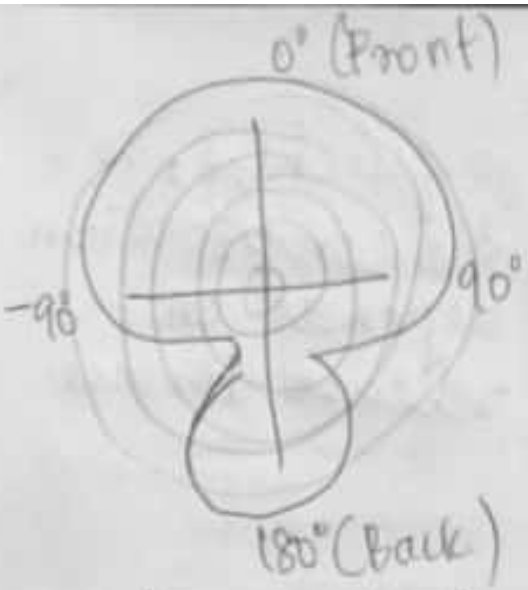
Polar pattern of MIC - It records the attenuation pattern of a MIC



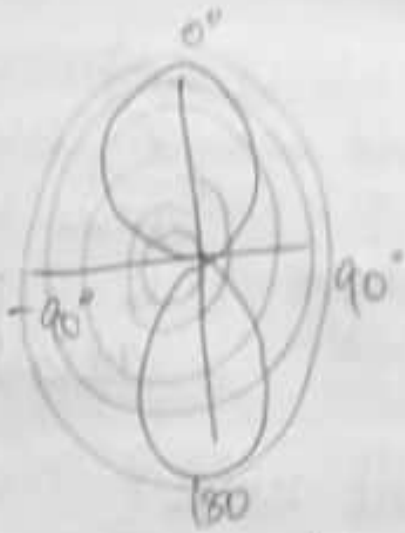
(i) Omnidirectional

(ii) Subcardiod (picks up from front, not so much from back)

(iii) Cardiod



Hypercardiod
(iv)



Bidirectional
(v)

(ii), (iii), (iv), (v) are all directional mic

↓
noise reduces, focus more on speech

The diagram is the polar pattern showing the amplification at the front (0° degree), rear (180° degree), right side (+90° degree), left side (-90° degree).

The area where the microphone system has higher sensitivity is called lobe.
Minimal sensitivity is called null.

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Dynamic Aspects of Compression :-

Loud & soft sounds come & go quickly, so how well the compression is applied depends on attack & release times.

↑ less gain — ↓ more gain

Why reduced noise?

* Hear speech better
* Reduce discomfort
* Speech intelligibility (noise masks speech)
* Reduce listening effort.
* Incorrect sound natural when noise is amplified (distortion)

So, to improve speech intelligibility → Directional microphone (picks up sound from front, reduces sound from back) ↑ SNR → Better speech intelligibility for effective comm.

Diff. B/w Noise & Speech (Noise Reduction Algorithms)

Normally signal is ...

Envelope - A curve which connects all peaks of the waveform.

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The waveform that is defined by the maxima of sinusoid.

No. of cycles per second ⇒ Modulation frequency → Usually less than original waveform
(envelope)
↳ forms a sort of signal of envelope

Distance b/w max. & min amplitude is called modulation depth.

If you connects the peaks of the signal → you get envelope

Q. How do you differentiate Noise from Speech?

phonemes, syllables, words, sentences

Speech consists of. We use modulation based technique.

Any syllable change, causes amplitude change. (So there is modulation)
↙ or modification or fluctuation.

Phoneme itself gives a modulation → more in no.
freq of 12Hz
Syllables - 5Hz
Words - 2.5Hz
Sentences - 0.3Hz. } =20Hz

For Noise, modulation freq is more than 20Hz.

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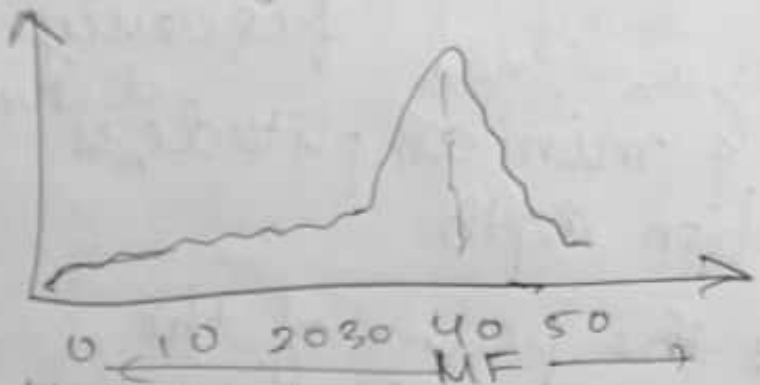
Noise Algorithm

→ Modulation based

→ Wiener filter.



Noise is 40
(greater than 30)

So, here noise is more

Processing → separating noise from speech.

Mic sends sounds to many filters/channels
(Low, Mid, High)

They take care of processing. Each channel gives SNR.

If speech > noise, it allows max gain.

If noise > speech, gain is reduced in that channel.

So all the channels together → SNR↑
more signal, less noise

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Modulation Based technique separates speech from noise in terms of modulation freq & modulation depth.

- **Wiener Filter** (to detect speech or separate speech from noise)

Signal → Speech + Noise.

This filter gets two info:-

↳ Noise (N) (goes few moments before speech because noise in the env.) → constantly its measured.

↳ Speech + Noise (S+N) → Power of Speech.

S + N − N = Speech (subtracting both you get power spectrum of speech.

Formula for determining SNR ÷

$$W(f) = \frac{S(f)}{S(f)+n(f)}$$

(Noise has to be less for a Wiener spectrum to increase)

Based on the power of speech & power of noise, wiener formula is calculated. This formula is applied to each channel.

Noise is less, Gain ↑. Depending on SNR gain is determined

Noise is more, Gain ↓ at that channel.

W(f) value ↑'s - for noise ↓

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Noise reduction algorithm has many advantages written before (comfort etc...)

listeners effort (Go Near, concentrate more, no multitasking, ask for repetitions)

they need to comfortably hear.

Impulse Noise (loud & sudden.) : Transient sounds. (sound of plates)

Wind noise : Sound of wind. (going by vehicles)

Feedback: possible in any amplication system.

What goes into mic, comes back through receiver goes into mic again)

(Feedback (cancellation/Reduction/Management)

Books: Dillon, Sandelaine.

HA differentiates noise from speech using characteristics.

Using modulation.

Modulation freq is lesser for noise than speech.

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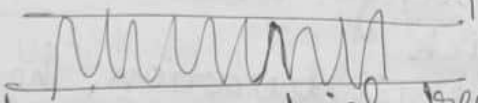
Original waveform will have higher frequency than the modulation frequency.
Original waveform has more peaks/cycles.
More cycles → More frequency. & vice versa.
Envelope → connects all multiple peaks to form one wave.
So, less cycles → less frequency.
Noise. Has no envelope.



has very high frequency compared to speech envelope.
Less than 20Hz → speech
more than 20Hz → Noise.
Modulation Based Technique: Based on modulation freq & depth → Modulation is done.

Weiner Filter & Modulation
↳ Both is used for constant noise.

Attack Time: 50ms. . X speech onset is similar impulse noise.
Release Time: 200ms
(always more than attack time)
A sound needs to occur atleast 50ms for compression to happen. Impulse noise has

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a ~~sudden~~ duration less than 50ms.
So compression cannot be applied.
Impulse noises are loud & sudden.
So we need a separate algorithm ~~needed~~ for impulse noise to detect it & remove it. This algorithm doesn't affect speech. (Regular compression cannot remove impulse noise)

Speech signal → longer time.
Noise → Quick.
Because of this diff. in duration; the algorithm can separate noise from speech.

Here, 2-8 channels is good enough to reduce high impulse noise.

If this algorithm is too long, the actual loud acoustic env will sound soft which is not natural.)

Wind noise: Noise made by wind breeze. (more prominent in 1st person)
↳ of low frequency.

So, we need a high pass filter that takes away the low frequency sounds.

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Wind hits 2 mic
This algorithm takes sound from both mic & correlates & see if the info is same or not. These two info. if are not correlated at all, so this signal is cancelled because wind noise has too many frequency & acoustic characteristics will not be same
So this wind noise going into both mic is not correlated, so it is cancelled out.
Speech → Has good correlation (input coming from 2 mic)

Mic Noise Reduction :-

Its working, so it has noise. → low freq → because in quiet env it can be heard causing discomfort
It causes a soft noise. (around 20 dB)
Bcuz HI person, may have some freq normal & some freq affected.
It uses a high pass filter. Anything below the cut off freq is attenuated. Above cut off freq it is passed.
× Reduction of Gain in the low frequencies. ×

Feedback System (in HA's → squealing)

I/P → Mic → Amplifier → Reciever → O/P → Person's ear

part of the O/P → Feedback loop.

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One particular sound gets amplified and a loop is formed. (Certain freq keep getting amplified)
At one point it stops, when there is a max. limit of reciever.
Amplified up to max. limit of the reciever.

↳ It has a limitation beyond which it can't give O/P

This loud sound which keeps getting amplified is only heard as squealing.

In olden days, they used to reduce the overall sound rather than reducing the ~~gain~~ freq causing squealing. So, the patient used to hear softly

Source (of sound escape)

↳ Vent
↳ Open fit (eartip/dome) } hole is present ↳ sound is constantly going out - stationary

↳ Gap b/w vent & ear canal → dynamic
↳ Dome & ear canal → dynamic (changing)

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When you eat/ chew, → only at that time, sound escape shape of EAC changes so intermittent gaps are formed.
(Gap b/w Vent & EAC)
(Bone & ear canal)

How can you reduce feedback/squealing?
Because sound from mic goes to frequency filters or channels, the gain is reduced at that particular filter/freq. So ∵ squealing is due to the amplification of a particular freq, it is eliminated.

Frequency lowering Technology.
Lowering high frequency sounds.
Basal part of cohlea are more frequently affected than apical region.
Basal part → High frequencies
↑
If affected → Dead region.
When a soft region given at dead region → cannot hear.
A very loud sound given stimulates

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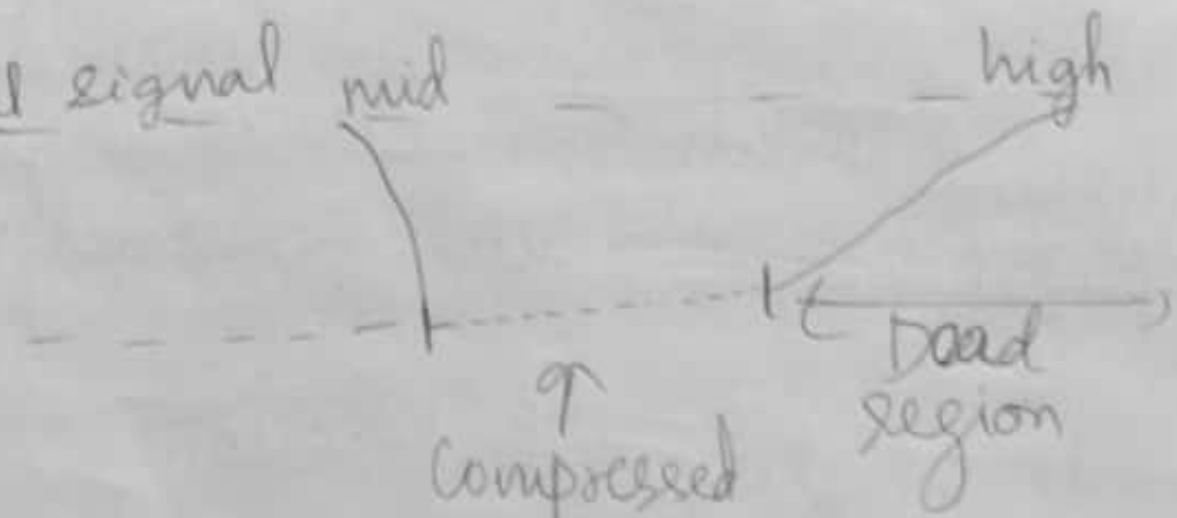
the neighbouring area and they can hear a ~~cattle~~ so they can't hear high frequency sounds like /s/, /z/ etc.
So we are giving them sounds in a region where the hair cells are alive/aligned. So frequency lowering technologies is used. It includes;

* Freq Compression.
* Transposition.

Frequency Compression - Squeeze mid & high frequency sounds in the adjacent area.



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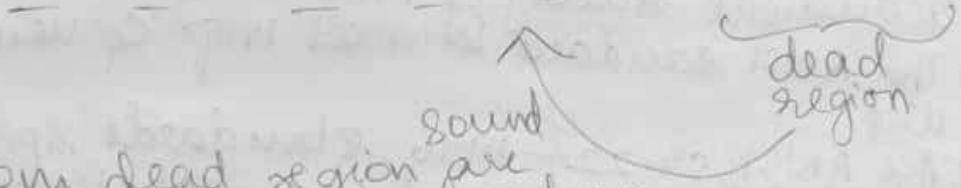
& Transposition (Otican HA) (high frequencies are replicated into better mid frequencies)

The sounds coming in the dead region are picked up and placed on top of each other in the adjacent areas of the cochlea.

Normal signal



Sound from dead region are picked up and placed on top of another in the adjacent area

Note:-

Digital Noise Reduction

Noise: steady intensity over time

Speech: intensity fluctuates rapidly over time.

DIR mics → objectively increase speech over noise

Noise Reduction → subjectively ↑ intelligibility ↑'s comfort in noise / reduce effort

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HA's can ~~either~~ be both directional / omnidirectional

noisy quiet

www.scribd.com (Electroacoustic measurements in HA)

Imp. EAC

ANSI gives specifications. IEC (International Electrotechnical Commission)

ANSI standards :-

- These standards have been developed under the ANSI standards committee method of procedure under the sponsorship of the acoustical society of America. (ASA)
- The ANSI standard is most imp & widely used
- The ANSI S3.22-1996 standards specify that tests be performed using pure tones. (2014 is the recent specification).

Frequency Accuracy: within ± 2%

Control Mic system: the press ure response level of the control mic system shall be constant within ± 1dB over the frequency range 200-5000 Hz

Coupler Types :-

ITE and canal instrument HA-1 direct across coupler. A vent must be sealed at the

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outer opening.
BTE instruments - HA-2 coupler/ ear mould, stimulator, with the tubal extension body
Work/ instruments - HA-2 Coupler/ ear mould stimulator without the tubal extension.

BTE, RIC — HA-2 Coupler

CIC, ITE, ITC - HA-1 coupler
(Custom HAIs)

Mic — Coupler — HA

Should be equally sensitive (200-5000 Hz)

Coupler Mic :-
Flat within ±1 dB over the range 200 to 5000 Hz calibrated within ±1 dB at any stated freq. b/w 250 & 1000 Hz within ±1 dB of true rms for noise signals.

Average time :-
≥ 0.5 sec, for internal noise measurements.

Battery Current :-
Accurate within ± 5% DC voltage drop ≤ mV, AC impedance ≤ 1 ohm over the range 20 - 5000 Hz.

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Ambient Temp.: 23°C ± 5° (73°F ± 9°)
Ambient relative humidity: 0% to 80%
Ambient atmospheric pressure: 760 mm of Hg (+35, -15) mm of Hg (or 101.3, ±5, -10 kpa)

Output – dB SPL (OSPL-90)
Gain – dB (Full on Gain)

Reference Test Position (when its 17dB less than the output - that position is reference test position)
Reduce Volume 17 dB from the max. output.
Gain ↓, output ↓.

Equivalent Input Noise Level: totally ~ 20 and 20 dB. + 3 dB (is accepted) to 23 dB.
It is beyond the ANSI specification if its not meeting the ANSI standards.

Test for AGC in HA's.
Change the HA's to no compression HA's. So it gives equal gain across all outputs. (Rest every thing)

Compression to max. use measure only

Attack time & release time — Input-Output Curve

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ANSI S3.22 (2003) Revision :- HFA - OSPL90 (±4dB)

FOG, RTG → 50 decibels are given. HFA - FOG (±5dB)

OSPL-90 → reference freq had max output (±3dB)

ISI (Indian Standards Institutions)

↳ Feb 1984. (Classification)

HFA's

mild moderate Strong Class.

(Based on OSPL90, FOG, HFA-FOG, Saturation Sound PL.)

Based on Substitution method (1 mic)

Freq. Response - 50, 60, 70, 80, 90 dB. (multiple intensities are given)

IEC Standards :-

RTest position - 15 dB.

at one freq → 2kHz, Varies the intensity b/w 55 to 90 dB from 90 dB/s

When it goes beyond the knee point, release time is checked

the attack time is checked.

ANSI

EQ Input Noise level (dB)

Tolerance : ±3dB.

How much it should be : Manufacturer.

We are checking it if it is working acc.

tolerance

THD : tolerances ±3% of the day 3% ; it can go upto +3% Max. 6%)

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to what they are saying it should be. So we are checking only based on the tolerance.

Document: ANSI S3.22 (2003)

FOG : Maximum Gain

Output & Gain tell us about the power of HA

Mild, Moderate, Severe & profound losses.

142/82 ⟶ profound loss.

MPO/FOG

~~[redacted]~~

HA - troubleshooting

(Whenever the HA has a problem, how to identify it & correct it?)

truesoundhac.com/hearing-aid-devices/troubleshooting-and-maintainence

If HA is not giving any sound/output it could be due to;

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↳ battery is dead (10/13/312/675) (Replace with new battery)

↳ wax blockage
(Use silica gel to absorb moisture. Use dehumidifier.

Battery costs 30 rupees.
(Change every 2-3 weeks, just open battery compartment)

↳ Squeating (whether the tip is inserted properly, & whether tip size match with the ear canal size).
If its loose, there'll be leakage → squeating.
keep replacing the mould because the ear canal grows.
Every 6 months - Young children.
Every 1½ years - adult

↳ Sound are weak/distant.
(due to wax, battery low)
moisture in the device

↳ HA buzzes but you cannot hear other sounds:-
(Problem in mic)

www.seattlehear.com/blog/
HA troubleshooting tips.

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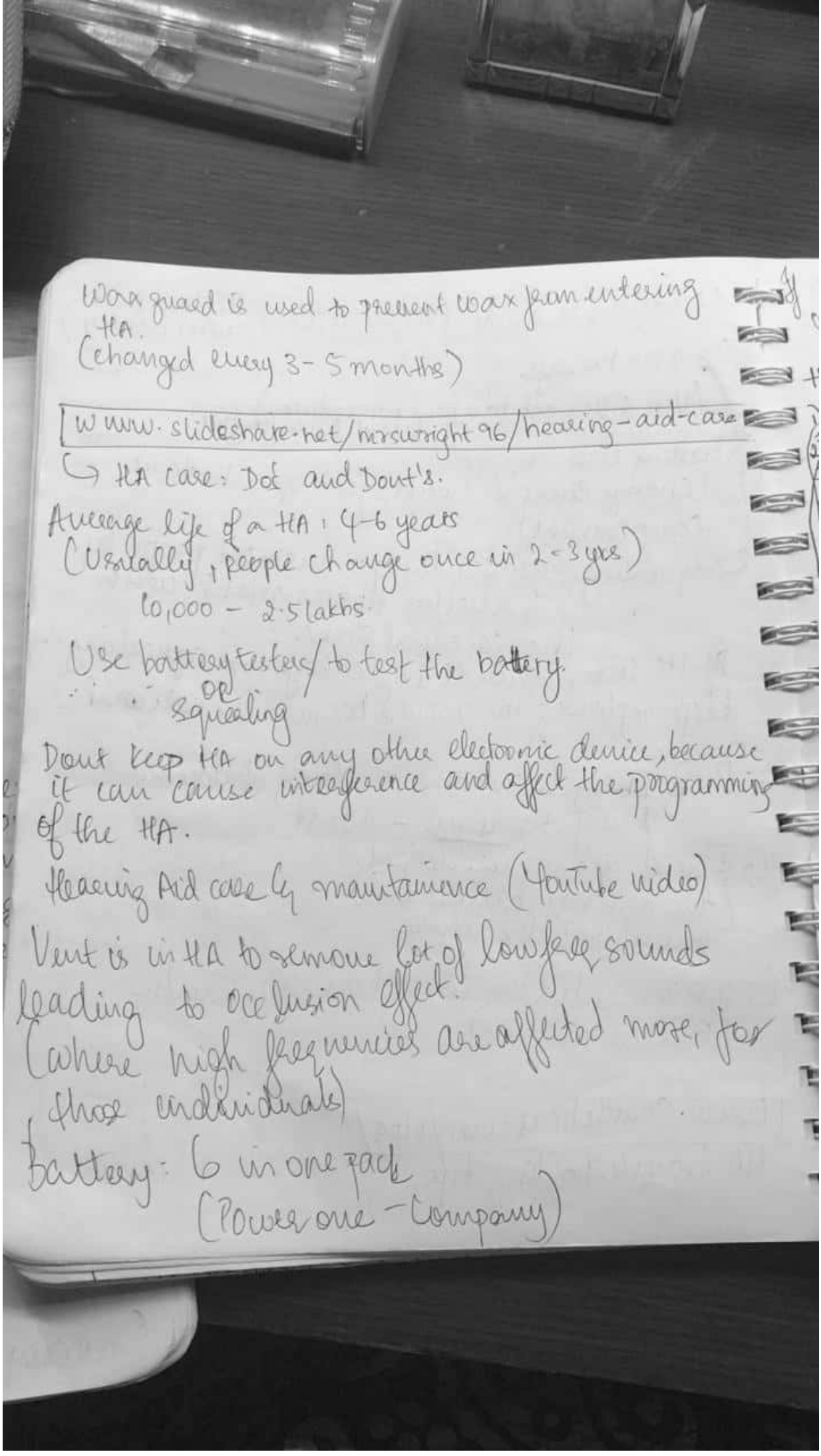

Wax guard is used to prevent wax from entering HA.
(changed every 3-5 months)

www.slideshare.net/mrswright96/hearing-aid-care

↳ HA care: Dot and Dont's.

Average life of a HA: 4-6 years
(Usually, people change once in 2-3 yrs)

10,000 - 2.5 lakhs

Use battery testers/ to test the battery.
or
squealing

Dont keep HA on any other electronic device, because it can cause interference and affect the programming of the HA.

Hearing Aid care & maintainence (Youtube video)

Vent is in HA to remove lot of low freq sounds leading to occlusion effect.
(where high frequencies are affected more, for those individuals)

Battery: 6 in one pack
(Power one - Company)

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If HA doesn't stop speaking even after closing sound outlet, it means there's an internal problem.

Hearing Aid Selection :-

1) Comparative Method — HA should be given based speech Audiometry (audibility, tolerance, performance with/without HA)
2) Prescriptive Method.

During war, many people had lost hearing.

Earlier 10 - 40 dB (slow growth of loudness)
70 - 90 dB (fast growth of loudness)
Earlier - linear aids.

→ Choose the HA which gives best performance in speech. (Compare & contrast) [speech scores with & without hearing aid]

→ In 1937, research was done. Which aspect of sound is giving the best benefit?

West in 1937 gave this method.

Audiogram → After the method

(Mirror image method)

Giving the mirror image of the audiogram by reducing the gain by half in respective frequencies. (Each frequency, formula is applied)

This method was popularised by Watson &

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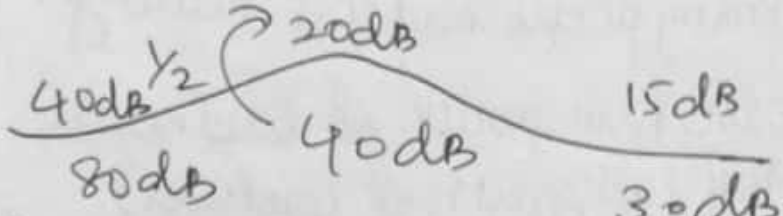
Eg Nutson in 1940. rule or ½ gain rule
When mirror image was applied to Severe-profound losses, it was very loud.
So, thats when loudness adjustment was thought about.



If after applying this rule, the target curve matches the prescriptive formula curve, the patient would be able to hear.

www.hearingreview.com/2003/11/essentials-of-hearingaid-selection-part2-its-in-the-numbers

Goal of prescriptive techniques has been to specify the performance characteristics of the individual patients loss

Lybarger (1944) → Half Gain Rule.

First linear formula was there. Later, non linear formula came.

Non-linear prescriptive formula - FIG 6, IHAFF, DSL, NAL-NL1

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Soft sounds → more gain
Loud sounds → Least gain.
Avg. conv. speech levels → 60-70 dB.
NAL-R (National Acoustic Lab-R) formula
↳ maximize speech intelligibility.
↳ bringing all frequencies in the same loudness range.

The POGO II Procedure
~~The~~ accepted (½ gain rule) and made some corrections to it for loss above 65dB.
upto 65dB - ½ gain rule
Beyond 65dB - Correction Factor
When UCL values changed, some CF was applied.

Non-linear

~~HAFF~~ protocol (loudness normalization method)
Soft sounds are perceived as soft.
Average → average.
Loud sounds → Loud.
(Compression)

Fig-6 (normalizes loudness) for input levels of 40 dB, 65 dB, 95 dB SPL.

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Finding an input intensity correctly for a person

DSL (i/o) → popular among children
uses REAG (Real Ear aided gain)
to provide users with audible, comfortable signals in each frequency region.

NAL-NL1 → doesn't normalize loudness.
Even if there is difference in loudness across frequencies, the aim is to maximize the speech intelligibility.
For different input levels, different targets.

Carhart Method

Goal was to select the HA that would provide max. benefit using the following measures;
1. Effective gain (using SRT stimuli)
2. Best Speech Discrimination (using PB-words)
3. Tolerance limits.
4. Efficiency in background noise.

SNR=5
Signal is 5 dB more than noise

SNR=-5
Noise is 5 dB more than signal.

SNR=0
Signal & noise is equal.

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linear formula: Same gain for all input sounds
Non-linear : Diff gain
↳ More accepted because we don't hear all sounds in the same way.

How will you choose HA's?

Hearing Aid Selection (family CIC/RIC/ITC/BTE)

What are the factors?

- CIC, BTE, RIC
- Ease of insertion (CIC, ITC > RIC > BTE)
- Ease of on-aid user control manipulation (BTE) ↳ Volume, battery compartment, changing batteries. CIC → hardest
- Invisibility (CIC > ITC > RIC > BTE)
- High gain & max-output (BTE) → Large HA offers more power
- Bandwidth & freq. response
- Directivity
- Insensitivity to wind noise Hanged dome or power dome.

Dome: Open, Closed, double, plus (too layers)

↓

give air tight seal, so no Leakage of sound. So no squeaker But low frequency sounds can escape So no RIC for hearing loss

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BTE with a ear mould
↳ profound HL unaided.

Ear canal is a high pass filter.
Resonant freq = 3500Hz for ear canal (most amplified in ear canal)

Concha: 5000Hz (resonant freq.)
So basically natural high freq. amplification.

So,
When you close ear canal, all high freq. are less, low freq. sounds are over amplified. This is ~~loss~~ occlusion effect. Thats why you hear voice (low freq.).

RIC has better freq. response at high freq. than BTE
↑ tip cuts off some high frequencies.

RIC (RITE ear) ; RITA (aid)
↑ receiver in ear. But if receiver is damaged, it can be replaced in clinic.
↳ receiver in the HA.

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Insensitivity to Wind Noise : CIC > RIC > BTE

inside the ear.

As the wind passes through the pinna & tragus, it causes turbulence which is picked up BTE.

Directivity: BTE > RIC > CIC

- Reliability → how long the HA can work. BTE > RIC > CIC
- Telephone Compatibility (BTE is best)
- Stethoscope Compatibility (BTE)
- Ease of cleaning (RITA > RIC > CIC)
- Avoidance of occlusion & feedback (RIC > BTE > CIC) for preventing occlusion
- Cost BTE > RIC > CIC occlusion
- Battery Size
- Factors in Combination → affects volume control, battery contact, battery compartment

To prevent wax to go into HA → wax guard

But moisture can go into HA

Life of CIC less than RIC & BTE

Waterproofing & nano coating → prevent wettage & moisture increases life of HA

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BTE - telecoil mode.

CIC → no telecoil mode. Mic is enough.
But there might be squealing
if you press against HA

Chronic ear infections/ → BTE is suitable
Ear discharge

feedback → for severe to profound.
BTE - lesser feedback. (gain is high, feedback also high, so more distance b/w mic & receiver)

Eg:-
low freq - normal ; high freq - loss
with HA → low freq amplified, high freq not so much
↳ occlusion effect

Making a hole called vent which will allow all the low freq sounds to come out preventing occlusion effect

More power, ↓ feedback → BTE. X.

CIC → more miniaturized circuitry
More worker labor → ↑ cost.
defects are more common in CIC, so cost ↑

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↓ battery size → ↓ power.
BTE's have longer than CIC
More features → more battery drain.

Pediatric → less sensitivity so they won't know if it falls down.
For profound loss, can't use CIC
Dexterity problems. (should know how to change batteries)

HA features

- Volume control
Compression, volume control, acceptable loudness level,
Disadvantages - Dexterity issues, ↑ed size of aid. (if patient want smaller aid)

- Wireless input (receiver)
Improves SNR using bluetooth technology
Eg: Watching TV, mobile phones.

- Directional Mic
Improves SNR, not possible in CIC/ITC
Disadvantages - Wind noise, Variable sound source

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- Compression Limiting (CL) Vs. Peak clipping

PC is preferred for following candidates.

- Patients with profound HL who need greater max. O/P
- For those who chose ITC over BTE with lower distortion.

Eg of CL - Multichannel compression or WDRC

- WDRC with high CR and low KPi is preferred for severe to profound HL.
- Advantages of CL - Listening comfort in noise and decreased need of VC.
  ↓
  Volume Control

HA features

↳ Volume control → Children don't know about it

Compression, VC, acceptable loudness level.

Disad - Dexterity issues, Need size of aid

↳ Wireless input (receiver) → power for distance.

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Improves SNR using Bluetooth technology
Eg: watching TV, mobile phones.

↳ Directional Microphones
Improves SNR, Not possible in CIC/ITC
Disad- Wind Noise, Variable sound source.

Peak clipping → happens in linear HA's.
(Gain remains constant
for soft, moderate, & loud sounds)

Fidelity → O/P resembling I/P
(no fidelity in linear HA's)

Compression → Multichannel
↳ WDRC

Linear HA's → sound very loud if not clear
& people are satisfied.
(for profound losses)

Non-linear HA's → more clear & comfortable.

For poor comfort → Compression Limiting.

↳ Adaptive Noise reduction
- Reduces gain in the frequency regions with

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Poor SNR
- Increases comfort (due to reduced noise)
- Not useful for high/low frequency ^HL. (Best in flat/gradually sloping loss) sloping

- ↳ Multiple memories (multiple programs)
  - There is a need of unique amplification characteristics for different listening situations.
  - Autoswitching b/w memories is possible with environment detectors in HA.
  - Not useful for high/low freq. HL

→ for different listening situations (Quiet, Speech in Noise, Noise, Speech in Quiet, Music, TV)
(diff. programs)

The HA will determine which listening env. it is, & the program changes accordingly.
If its too noisy, the directivity of the mic gets activated.

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Feedback Management

Squealing is common in following conditions
- Profound HL (∵ gain is ↑, more sound into ear canal → more sound left in the ear canal → more chances of sound coming out)
- High freq HL
- Open canal HL
- Using Telephone

Feedback cancellation is imp. for such conditions. However it will reduce the quality of music perception.

→ Buzz mostly high freq. sounds escape
Low freq. sounds generally travelling through BC mode as opposed to high freq. sounds which travel through AC mode / when you need more gain at high freq. sounds → more escape

→ Through hole, sound escapes

→ When sound escapes from mic, it hits the telephone & gets reflected back. So it causes squealing.

The mic will recognise the sound that has come twice. (after once). An opp. sound (opp. in phase) is created which

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will cancel the sound causing squealing

Data Loging:-

When you connect HA to software, you'll know how much the HA is on/being used.

It tells us in which listening situation also the HA has been used.

If patient gives it for servicing, we can see if the HA has been used or not.

How much the HA has been used from the time of purchase
↳ Data loging.

Factors on how to choose HA?

- Style of HA
- Features of HA
- Attitude ↳ motivation of client

Pg 118 to 120 (Dillon)

Feedback HA's. (224)

1. What is feedback (oscillation)
2. What is the mechanism

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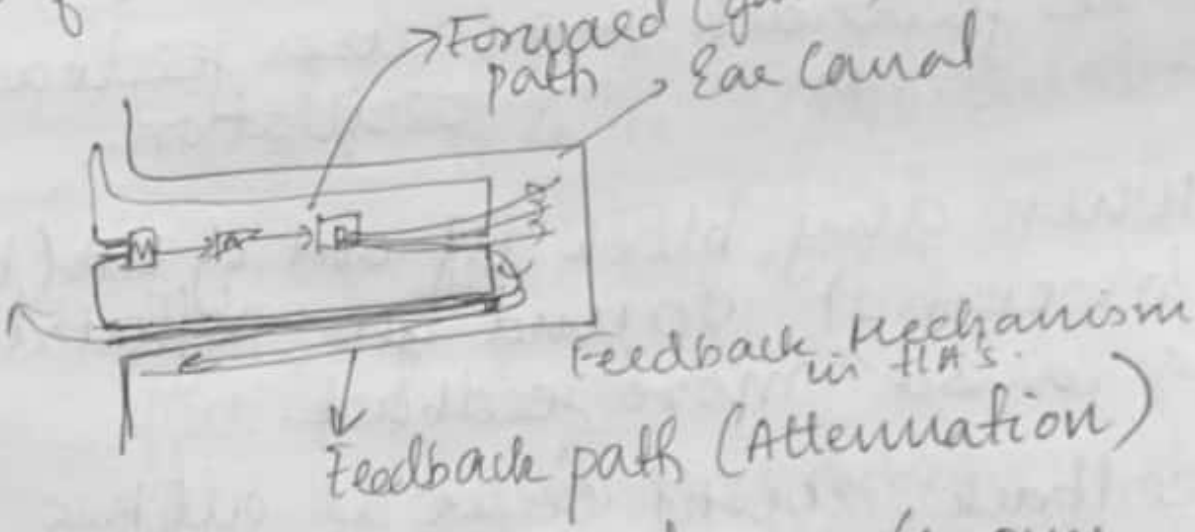
3) How it affects the sound quality?
4) How feedback is cancelled?



The sound gets amplified over & over until the MPO/ max. limit of receiver
Feedback is also known as feedback oscillation.

Compression, (OR) Peak clipping → Linear HA.
Non-linear HA → more, better way to control feedback (Better feedback mechanism

+ve feedback - Direct input sound into mic & sound coming from receiver into mic. If both are in same phase, the sound gets amplified more & effective gain is.

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-ve feedback → both sounds are opp. in phase, thus effective gain ↓'s.

-ve feedback → causes feedback oscillation.

When any block infront of HA (hand blocking), sound gets reflected more & causes more feedback

Feedback doesn't occur at all freq. (mainly in high frequencies).

Sound Quality :-

→ it'll be noisy (↓ speech intelligibility)

→ If the person has HL in the freq. where feedback occurs, though he won't hear, others will hear & it'll cause embarrasment.

How to reduce?

Reduce Gain where feedback occurs (freq)

↳ Channels (Using multichannels) (12/16)

less channels → most freq.'s gain will reduce, so ↓ speech intelligibility.

reduce gain only those freq. where there is feedback.

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feedback occurs only at high freq
Even after an input sound of high freq stops going inside, the feedback continuous causing a ringing sensation/ sound. This is Suboscillatory feedback. So, we need to reduce more gain in that particular freq where there is ringing sound.

Effect of feedback on sound quality

We can reduce it by knowing the freq where feedback is occuring and phase cancellation (reversing the phase of another signal by putting a tag on a signal that already gone through HA)

Pg 122, 123 (Troubleshooting)
Pg 124 (To control feedback)
Diagrams for feedback (Pg 124, Notebook Diag)
125 → Feedback mechanism in HA
↳ Causes of feedback in BTE/ITC (Diag)

Pg. 337 - 349 (factors affecting selection of HA's)

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Gain & Insertion gain Methods:-

One first step in the real ear gain measure is to determine the unaided gain or response curve (REUG or REUR). The next step is to measure the amplified gain or response curve and then subtract the unaided curve. The result is the aid's insertion gain. It is the curve of the gain that the aid gives the patient over and above the unaided condition. The insertion gain can also be calculated by using a normal unaided gain curve. This approach works if the patient physically has a nearly normal ear. the normal or average ear response.

The later or often recommended for the hearing impaired population although the nal system also accomodates this class of patients based on signal spectra and knee points and some of the formulae are set as (default) as a secret patient upto less than 1m away from the test ear, others use a closer spacing.

Communication Profile for the hearing impaired: CPHI

The CPHI provides a systematic and comprehensive assessment of a broad range of

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Canine problems - It was initially standardized on active duty service members at Walter Reed Army Medical Center. Later normative studies in

REM (Real Ear Measures)

HA output matching with target curve
↳ programming

based on audiogram

given by the formula

Target curve can be

Gain curve (mostly used)

Output curve

The target curve matched with HA output varies acc to patients. This is done by REM. After putting into patient's ear, the amplification may vary.

Helps us to see amplification at the level of EAC → REM

Lot of variations in ear canal size, shape, volume. Thus varies acc to diff people.

The target curve is standard, not acc to ear canal variation. So we need to check the gain acc to level of the ear canal.

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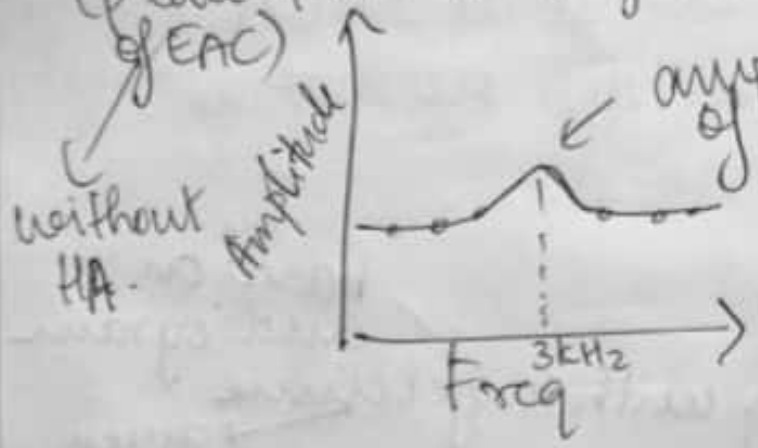
① REUR (Real Ear Unaided Response SPL) → first measure done
(place probe mic, give sound, & see gain at end of EAC)



Ear canal Resonance - In some freq. (resonant frequencies), the ear canal amplifies the sound.

Concha - 5000 Hz
EAC - 3000 Hz

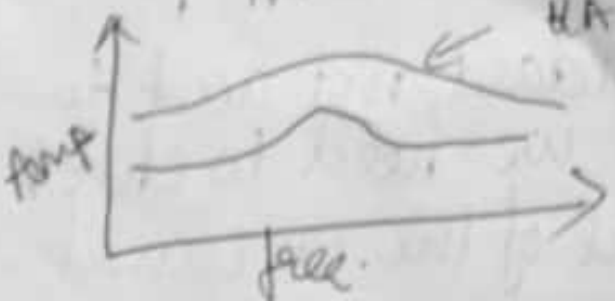
High freq are amplified in adults due to ear canal resonance.

If you take response/gain at each freq. or any freq. it is called Real Ear Unaided gain.

Curve is called REUR.

② REAR

Fit a HA, put it on.



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Levelling :-
All the sounds must be flat / levelled before testing is done. So that later on we know which freq. is amplified.

reference mic → outside ear
probe mic → inside EAC.

* Make sure HA doesn't cross comfortable limits.
* Target curve given by various prescriptive formulas must match HA output

Record results and save it. When there is problem, we can record again & see if there is any change. You need to show proof that the HA output is changing across time.

REDD.
Children are not cooperative to measure soft/med/loud inputs
HA company gives response for 2cc coupler.
But this varies. Infants can't cooperate.
We need to know the diff. b/w the ear & the

Priyanka Sood

Gain at particular freq → REAG.

after insertion

(3) REIG (Real Ear Insertion Gain)

The difference b/w 2 curves (REUR & REAR)

↳ How much gain is happening at the level of EAC

REIG is only the target gain. It is an estimate of the gain at EAC. So we can verify that by keeping a mic at the EAC and seeing if it matches with the software. Usually 80% of time it may not match, because every person's EAC varies acc. to size, shape, volume. The target curve gives values based on the average size, shape & volume not acc. to every individual

REIG should match with the target gain by various prescriptive formulas.

If not, fine tune the output of HA.

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Inserts give sound. Probe mic measures the gain across frequencies within the ear

Subtracting the Real Ear Value – Coupler Value (gives REDS value) — Check separately for both

The manufacturer has an (OR) sometimes coupler value for a given group. So you can use it & just measure real ear value

HA benefit is there / not → Func. Gain (Behavioural Respon

The diff. in gain with & without HA.

(To verify benefit of HA → Func. Gain & verify REM)

HA output → EAC.

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